

Fill In the Details below(IP is that of the Onestream box

Step 1.1 of 5

Internet Telephony Service Provider

Provider Name:

Enable Provider:

Domain Name:

Provider Registrar

Use Registrar:

IP Address / Host name:

Port:

Reregistration Interval at Provider (sec)

Provider Proxy

IP Address / Host name:

Port:

Provider Outbound Proxy

Use Outbound Proxy:

IP Address / Host name:

Port:

Provider STUN

IP Address / Host name:

Click OK & Next

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Internet Telephony Station for TelecomFM

Internet telephony station:

Authorization name:

Password:

Confirm Password:

Call number type

Internet Telephony Phone Number

Internet telephony system phone number

Internet Telephony Phone Numbers	
<input type="button" value="Add"/>	<input type="text"/>

Enter here all 'Internet Telephony Phone Numbers' provided by your network provider.
During configuration of the stations, you can assign the individual numbers to them.

Fill In the Details (all these are just to register the SIP trunk and can be changed)

Internet telephone station – 300

Authorised Name – 300

Password - 12345678

Confirm Password – 12345678

Add a Phone number I used 123456 again this is so traffic can be passed through the trunk Click Add

Click OK & Next

On the Next Screen Select the new Station you created and change the internal call number to either a phone or a group, depending on how you want to route the call

Click OK & Next

Click OK & Next

Step 2 of 5

Settings for Internet Telephony

Simultaneous Internet Calls

Please enter in field 'Upstream up to (Kbit/sec)' the Upstream of your Internet connection communicated by your Provider. You have typed in **Upstream up to (Kbps) = 256** in the 'Change Feature --> Internet Telephony' Assistant. This upstream allows you to conduct up to 2 Internet phone calls simultaneously for each active Internet Telephony Service Provider. If the call quality deteriorates due to network load, you need to reduce this number of simultaneous calls.

Upstream up to (Kbps):

Number of Simultaneous Internet Calls:

Select the Amount of channels you want, I have selected 2 as the unit is a 2 SIM Unit

Click OK & Next

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Special phone numbers

Note:
Emergency calls should always be built up with ISDN or Analog Trunk for safety reasons.
Please make sure that all special call numbers are supported by the selected provider without fail.

Special phone number	Dialed digits	Dial over Provider
FAX/Modem		
1	<input type="text" value="9C112"/>	<input type="text" value="ISDN"/>
2	<input type="text" value="9C999"/>	<input type="text" value="ISDN"/>
3	<input type="text" value="9C07XXXXXXXX"/>	<input type="text" value="TelecomFM"/>

Here we have how to route the call, from the drop down select the provider TelecomFM and the Dialed Digits 9C07XXXXXXXXX this tells it to route calls started with 07 over the GSM Gateway

Click OK & Next

Click Next

Click Finish

This is now the SIP Trunk Set up on the Openscape MX

Configuring the Onestream G

Power on the OnestreamG

Run the One Stream Scanner and locate the Unit

Log in to it with password 12345678

The screenshot shows the Onestream G web interface. On the left is a blue navigation menu with options: Home/Status, Groups, Routes, Advanced, SMS, SMS Options, Security, LAN Settings, Time/Date, Load/Save Config, Update Firmware, Diagnostics, Restart, and Logout. The main content area is titled 'System Information' and lists: Uptime (49 min), LAN IP Address (172.22.22.200), MAC Address (00:50:C2:60:73:23), Serial Number (9498310), IMEI Module 1 (352022000874782 (MC55)), IMEI Module 2 (352022000874733 (MC55)), Firmware Version (2.6.3663), Firmware Date (Wed 16 Mar 2011 12:33:24 GMT), Config Version (2.6.3663), Config Date (Wed 16 Mar 2011 12:33:24 GMT), and System Date & Time (Fri, 25 Feb 2011 04:51:09). Below this is the 'Status' section with icons for POWER (ON), GSM (NO SIM), IP (IDLE), FXS (IDLE), FXO (IDLE), SIGNAL (3), and WARNING (0). A 'CHANNEL STATUS' table shows GSM 1 as IDLE on O2 - UK and GSM 2 as NO SIM. The 'SIP Networks' section at the bottom states 'No SIP Networks registered. Select "Groups" page to add a SIP Network.'

Click the LAN Settings on the Left

The screenshot shows the 'LAN Settings' page in the Onestream G web interface. The left navigation menu is the same as in the previous screenshot, with 'LAN Settings' highlighted. The main content area is titled 'LAN Settings' and features a form with radio buttons for 'DHCP' (unselected) and 'Static' (selected). The form fields are: IP Address (172.22.22.200), Subnet Mask (255.255.255.0), Gateway Address (172.22.22.254), Primary DNS (8.8.8.8), Secondary DNS (4.2.2.2), Domain, and Hostname (OneStream_9498310). 'Save' and 'Cancel' buttons are at the bottom.

change it so it is on a Static IP address and Click Save

Click Groups on the Left

Then Add Group Sip Extension

The screenshot shows the 'Edit Group "MXSIPEXTENSION"' configuration page. On the left is a navigation menu with options like Home/Status, Groups, Routes, Advanced, SMS, SMS Options, Security, LAN Settings, Time/Data, Load/Save Config, Update Firmware, Diagnostics, Restart, and Logout. The main content area includes:

- Interface Type: SIP Extension (dropdown)
- Buttons: Save, Cancel
- Name: MXSIPEXTENSION (text input)
- SIP Extension section:
 - Options: SIP Port (default 5060) (text input)
 - NAT Traversal: Disable (dropdown)
 - Call Limit (0=unlimited): 0 (text input)
- Extensions section:
 - Username: 300 (text input)
 - Password: [masked] (password input)
 - Buttons: New Extension, Save, Cancel
- Link: Show Advanced Options

Add the username you created in the MX and the Password then click save.

Click Routes

The screenshot shows the 'Routes' configuration page. On the left is the same navigation menu as in the previous screenshot. The main content area includes:

- Buttons: Add Route
- Table of routes:

Route	From Group	Dest. Addr.	Orig. Addr.	To Group	Mod. Dest.	Mod. Orig.	ACT	CB	Split/Fail	Edit	Delete
GSM -> MX	GSM	s	-	MXSIPEXTENSION	123456	-	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		
MX -> GSM	MXSIPEXTENSION	07?	-	GSM	-	-	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		

And Add routes to the system

You Need to create 2 routes, One from the Phone system to the Onestream and one from the one stream to the Phone system. On the Route from GSM->MX you need to modify the destination address so the system knows how to handle the incoming call, this is where you put the MSN of 123456 this allows the sytem to route the call to the phone.

To check the trunk is registered in the MX click Service Centre ->Diagnostics ->Status ITSP Status

Service Center - Diagnostics - Status						
Network Status	Station Status	Dialup Network Status	ITSP Status	VPN Status	Dial Plan	Overview of IP Addresses
Step 1 of 1						
Status for the Internet Telephony Service Provider (ITSP)						
	Provider		User			
	OPAL	Disabled				
	OpenIP	Disabled				
	Priority Telecom	Disabled				
	Sipgate	Disabled				
	Sipgate Trunking	Disabled				
	Skype Connect	Disabled				
	SoTel	Disabled				
	SoTel with register	Disabled				
	Tele2 NL-ASD	Disabled				
	Tele2 NL-RT	Disabled				
	TelecomFM	Enabled	300		registered	

Your SIP trunk should be displayed as Green

In the Onestream box Click Home/Status on the left hand side and

The screenshot shows the Onestream status page. On the left is a navigation menu with 'Home/Status' selected. The main content area displays system information (Firmware Date, Config Version, etc.), a status bar with icons for Power, GSM, IP, FXS, FXO, Signal, and Warning. Below this is a 'SIP Networks' section with a message: 'No SIP Networks registered. Select "Groups" page to add a SIP Network. SIP Network Groups that do not require registration will not be displayed here.' The 'SIP Extensions' table shows:

Username	IP Address	Status
300	172.22.22.250	Registered

The SIP Extension should have a Status of registered.

To test dial a mobile number and it will display the incoming number as the SIM in the Onestream

And if you have configured the MSN to ring to an extension when you dial the mobile back it will route the call through to it.